JVP™

AnTares Voice Processor
for TDM

USER'S MANUAL

ANTARES
REALLY COOL STUFF FOR MAKING MUSIC

JVP - AnTares Voice Processor™

Antares Audio Technologies
464 Monterey Ave, 2nd Floor
Los Gatos, CA, 95030
(408) 399-0008, (888) 332-2636
web: www.antaresTech.com

JVP License Agreement

Carefully read all the terms and conditions of this license agreement. If you do not agree with the terms and conditions of this Agreement, notifying Antares Audio Technologies and destroying all copies of the manual, and erasing JVP from all machine-readable media, whether on-line or on archival copies.

Software License Agreement

Antares Audio Technologies grants you a non-transferable, non-exclusive license to use JVP, under the terms and conditions stated in this agreement. Use of JVP indicates your agreement to the following terms and conditions.

License

You may:

1. use JVP on only one computer at a time;
2. physically transfer the program from one computer to another, provided that the program is used on only one computer at a time.

You may not:

1. make copies of JVP or of the user manual in whole or in part except as expressly provided for in this agreement. Your right to copy JVP and the user manual is limited by copyright law. Making copies, verbal or media translations, adaptations, derivative works, or telecommunication data transmission of JVP without prior written authorization of Antares Audio Technologies, is prohibited by law and constitutes a punishable violation of the law.
2. make alteration or modifications to JVP (or any copy) or disassemble or de-compile JVP (or any copy), or attempt to discover the source code of JVP.
3. sub license, lease, lend, rent, or grant other rights in all or any portion of JVP (or any copy) to others.

Term of the Agreement

This agreement is effective until terminated by you or Antares Audio Technologies. You may terminate the agreement at any time by notifying Antares Audio Technologies and destroying all copies of the manual, and erasing JVP from all machine-readable media, whether on-line or on archival copies.

In the event of breach of any of the terms of this agreement, you shall pay the attorney's fees of Antares Audio Technologies that are reasonably necessary to enforce the agreement plus resulting damages.

Limited Warranty and Disclaimer

JVP AND ACCOMPANYING MATERIALS ARE PROVIDED “AS IS” WITHOUT WARRANTY OF ANY KIND, EITHER EXPRESS OR IMPLIED, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE.

Antares Audio Technologies does not warrant that the functions contained in the program will meet your requirements. The entire risk as to the use, quality, and performance of JVP is with you. Antares Audio Technologies warrants the diskettes on which the program is furnished to be free from defects in materials and workmanship under normal use for a period of ninety (90) days from the day of delivery to you as evidenced by a copy of your receipt. AnTares’ entire liability and your exclusive remedy as to the diskettes will be the replacement of the diskettes.

SOME JURISDICTIONS DO NOT ALLOW LIMITATIONS ON HOW LONG AN IMPLIED WARRANTY LASTS, SO THE ABOVE LIMITATION MAY NOT APPLY TO YOU. THIS WARRANTY GIVES YOU SPECIFIC LEGAL RIGHTS. YOU MAY ALSO HAVE OTHER RIGHTS WHICH VARY FROM JURISDICTION TO JURISDICTION.

Limitation of Liability

IN NO EVENT WILL ANTARES AUDIO TECHNOLOGIES BE LIABLE FOR ANY DAMAGES, INCLUDING LOSS OF DATA, LOST PROFITS OR OTHER SPECIAL, INCIDENTAL, CONSEQUENTIAL OR INDIRECT DAMAGES ARISING FROM THE USE OF JVP OR ACCOMPANYING MATERIALS. THIS LIMITATION WILL APPLY EVEN IF ANTARES AUDIO TECHNOLOGIES OR ITS AUTHORIZED AGENT HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGE. YOU ACKNOWLEDGE THAT THE LICENSE FEE REFLECTS THIS ALLOCATION OF RISK. SOME JURISDICTIONS DO NOT ALLOW LIMITATION OR EXCLUSION OF LIABILITY FOR INCIDENTAL OR CONSEQUENTIAL DAMAGES, SO THE ABOVE LIMITATION MAY NOT APPLY TO YOU.

Welcome!

I would like to extend my congratulations to you on purchasing the AnTares Voice Processor. With your purchase of JVP, you have begun a relationship with my company which I hope will be long and gratifying.

As a registered user of JVP, you are entitled to notification of software upgrades, technical support, and to special introductory offers on upcoming products. We will be in contact with you to announce new opportunities and to solicit your feedback.

At Antares Audio Technologies, we are committed to excellence in service, quality, and technology innovation. You can count on us to listen to you and to keep our promises to you.

Andy Hildebrand, Ph.D.

andy@antaresTech.com

Table of Contents

CHAPTER 1

Getting Started

The AnTares Voice Processor is a unique software plug-in for TDM well suited to almost any recording, broadcast, and post production application.

JVP's functions include a de-esser, a compressor with a downward expanding gate, a parametric EQ, and a multi-tap delay all presented in a familiar user interface.

While JVP works brilliantly on vocal enhancement, its unique set of digital tools also makes voices, instruments, sound effects, and complete mixes sparkle with rich detail. Designed by musicians for musicians, JVP is an unbeatable combination of raw processing power and extraordinary value.

How To Use This Manual

JVP has a transparent user interface and is extraordinarily easy to use. However, some functions of JVP may not be immediately obvious. We strongly recommend that you read chapters 2 and 3 of the manual to take full advantage of the quality and control that JVP makes possible.

We assume that you already know how to operate ProTools/TDM. If you have questions about this, refer to your ProTools manual or call Digidesign for tech support.

For Those Who Hate To Read Manuals…

We recommend that you leave the manual within reach when first learning JVP. When you come across something that you don't understand, take a few moments to look it up in the Index and read about it. The time invested will be well rewarded.

The Contents Of The Manual

Chapter 1: Getting Started

This chapter explains what you'll need to get started with JVP, communicating with AnTares Audio Technologies, and how to use the manual.

Chapter 2: Introducing JVP

This chapter explains the scope of JVP's functions. Basic concepts about de-essing, dynamics processing, EQ, and delay effects are also discussed. The user interface is explained.

Chapter 3: Using JVP

This chapter guides you through the operating methods and controls of all the tools in JVP. We recommend this chapter as "must" reading for everyone.

Software Note

A Digidesign TDM system is requirements for running JVP.

Owner Registration

Your purchase of JVP entitles you to technical support, special introductory offers on new products from AnTares Audio Technologies, and notification of software updates.

Please fill out and return the Product Registration Card. The information on the card will allow us to communicate more effectively with you and will enable us to serve you better in the future.

Installing JVP

To install JVP, double click the installer icon. Information about the authorization process call be found in the installed read-me file.

Technical Support

If you have some problem using JVP that can't be solved by reading the manual, call technical support at (888) 332-2636, or (408) 399-0008 Monday through Friday between 9 AM to 5 PM Pacific Standard Time.

Also, you might find what you want at our web page:

www.antaresTech.com

You can also e-mail:

techsupport@antaresTech.com

CHAPTER 2

Introducing JVP

This chapter introduces you to JVP. Topics covered are basic concepts in compression, expansion, gating, de-essing, parametric equalization, and delay effects. Also covered are JVP's basic architecture and user interface.

JVP is designed to be used in any editing, sweetening, or mastering application. Since it incorporates four of the most useful digital signal processes into one tool, you will probably use it more than any other DSP plug-in inside the TDM environment.

JVP is particularly suited to working with solo tracks. Voices, winds, guitar, bass, and keyboards can be dramatically altered and enhanced using JVP's tools. Since JVP uses the most advanced DSP technologies currently available, the quality of the results it produces is unsurpassed.

JVP can also be used to great advantage in mastering stereo files. Its compressor/gate and EQ operate simultaneously in stereo and offer the best possible quality and flexibility.

Understanding Compression

Next to reverb, compression is probably the most important signal process used in today's studios. Simply put, compression reduces the *dynamic range* of a signal. That is, it reduces the difference in loudness between the loudest and quietest parts of a piece of music. Another way to think about this is that the compressor is acting as an automatic fader which fades down when the music gets loud and fades back up when the music gets soft.

Why reduce the dynamic range? Consider mixing a vocal into a pop music bed. Typically, pop music has a relatively consistent level of loudness. If an uncom-

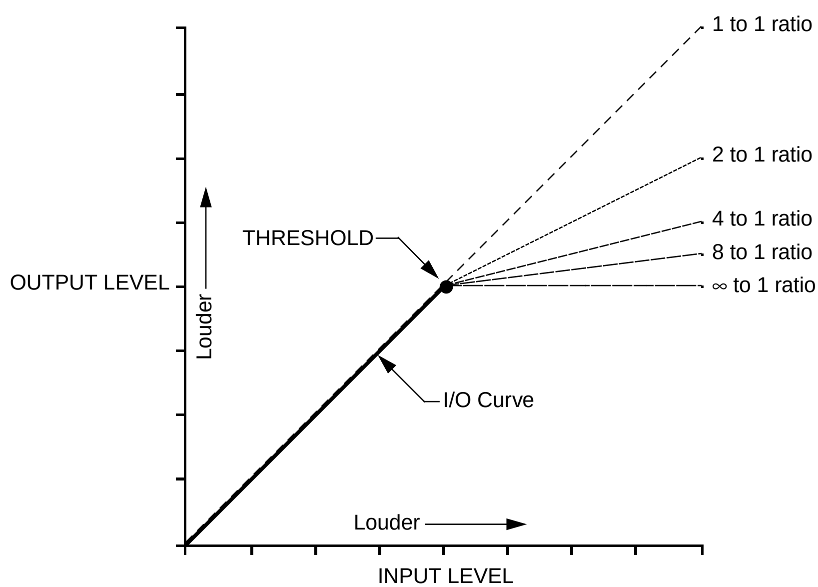
pressed vocal track is added to a typical pop mix, certain loudly sung words or syllables would be very obtrusive, while quieter phrases would be buried underneath the instrumental texture. This is because the difference between the loudest and softest sounds in the vocal, its dynamic range, is very large. This same problem occurs for any instrument which had a dynamic range larger than the music bed into which it is being mixed.

By using a compressor to decrease the dynamic range of the vocal, the softer sounds are increased in loudness and the loudest sounds are reduced in loudness, tending to even out the overall level of the track. This makes the vocal track sound generally louder and more distinct, and therefore, easier to hear in the mix.

Ratio And Threshold

How is compression measured? What is a little compression and what is a lot of compression? The concept called *compression ratio* is the measure of how much the dynamic range is compressed.

The graph shown below represents the relationship between the input level



of the signal and the output level of the signal after compression. Notice that the curve has a breakpoint called a *threshold*. All standard compressors use a threshold. Signals that are louder than the threshold are processed

(reduced in level) while those softer than the threshold are unchanged.

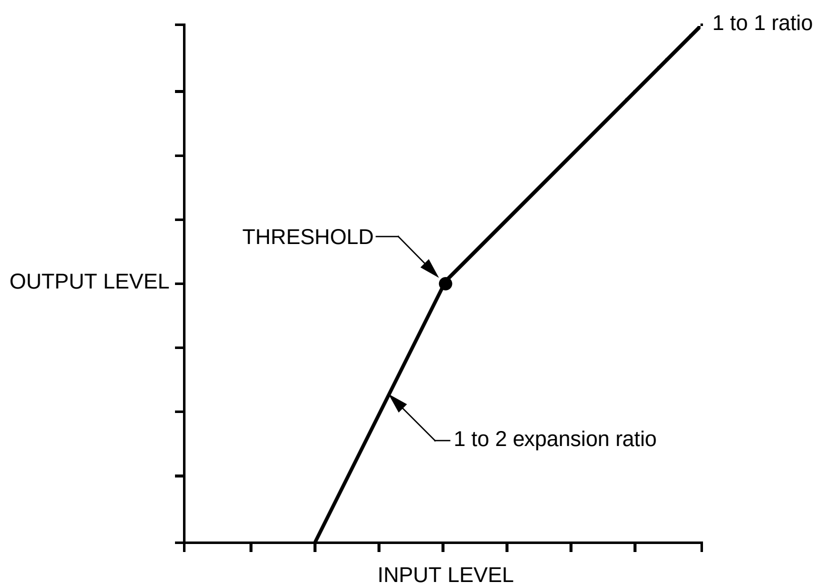
As the input signal exceeds the threshold, *gain reduction* (reduction in loudness) is applied. The amount of gain reduction that is applied depends on the compression ratio. The higher the compression ratio, the more gain reduction is applied to the signal.

The graph shows the relationship between compression ratio and gain reduction. Examine the 2 to 1 ratio curve. For signals above the threshold, this curve transforms a range of loudness 2 units large into a range of loudness one unit large. Examine the ∞ to 1 curve. This curve transforms all sounds above the threshold to the same loudness. Dynamics processors which have this sort of curve are called limiters.

Dynamic Expansion and Gating

Sometimes, it is desirable to increase the difference between the quietest signal and the noise in a recording by using a downward expander. A typical application would be eliminating room noises and breath sounds that can be heard in between the phrases of a recorded vocal part.

The graph below shows the curve for a downward expander. Notice that above the



threshold, the curve follows a 1 to 1 ratio. For each unit of input change below the threshold the output changes by two units. This is called a 1 to 2 expansion ratio.

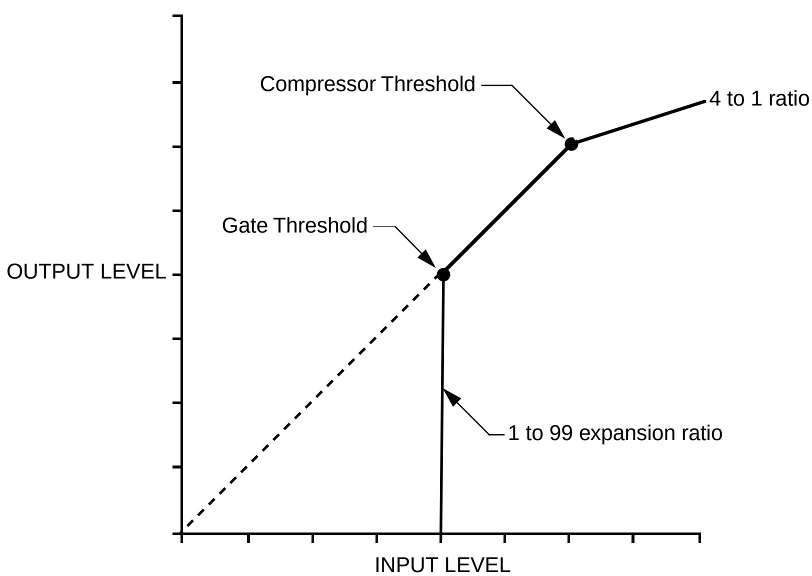
As the input signal drops below the threshold, its output level drops at twice the rate it would using a 1 to 1 ratio. In effect, sounds below the expander threshold are "faded out" more quickly than they would be normally. This effect can be exaggerated into what is normally called a gate.

When expanders have ratios higher than 1 to 10, sounds below the threshold are faded out very rapidly. This effect is called gating and can sound very abrupt. Adjusting the gate ratio can smooth out the abrupt change. The graph below shows the input/output curve for a typical gate.

Sounds that are louder than the threshold get "through the gate". Sounds that are below the threshold are not heard. Gates can be used to great effect in processing drum tracks where sounds from the other instruments in the drum set leak through the mike of the instrument being recorded. Gates are also used frequently to "gate off" a reverb tail or the ringing from an insufficiently damped drum head.

Compression And Expansion Combined

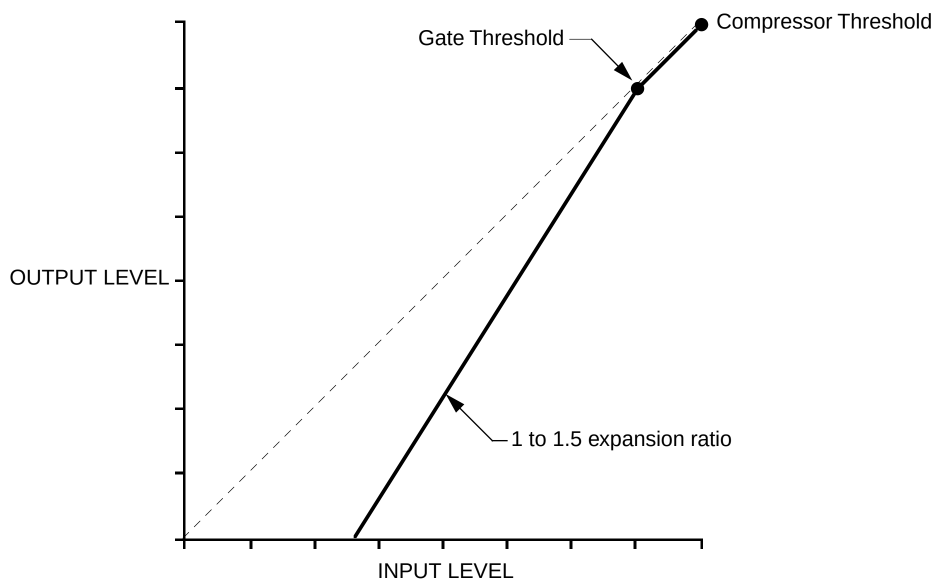
JVP allows you to use both compression and expansion simultaneously. This ability is useful in taming the typical problems that arise when processing vocal tracks. The graph below illustrates the use of compression with a downward expanding gate.



Using this setting, sounds above the compressor threshold will be compressed at a 4 to 1 ratio. Sounds below the compressor threshold but above the gate threshold will not be changed. Sounds below the gate threshold will be gated out completely.

Used on a vocal track, this setting will compress only hot peaks in the voice, while gating out the room sounds, mike stand sounds, and breath noises in the track. Precisely what gets compressed and gated is a function of the compressor and gate threshold settings.

The graph below shows a dynamic expander. In this application, the gate threshold and ratio are set to gently expand the program material at a 1.5 to 1 ratio. The compressor ratio is set to 1 to 1. The setting is useful for repairing over-compressed material or for adding some punch to drums or other percussive sounds.

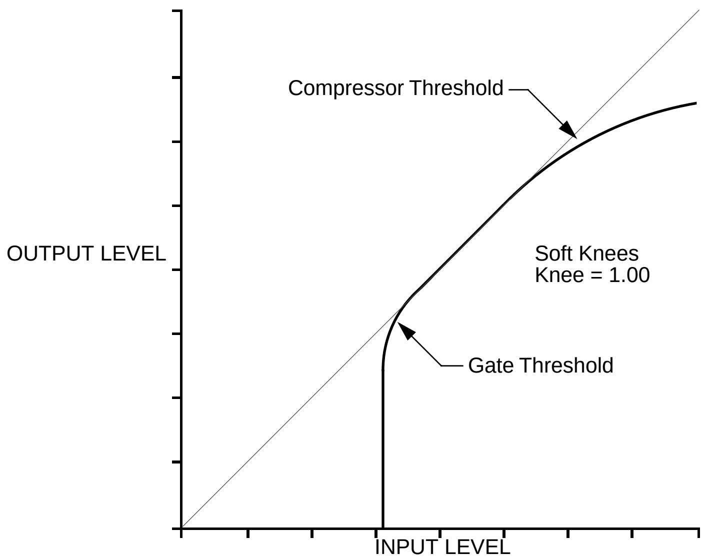


Hard Knee/Soft Knee

The graphs shown above have what are described as "hard knees" in their gain curves. This means that as the signal passes through the threshold, the gain reduction it receives will change abruptly. In settings where the com-

pression or expansion ratios have high values, the abrupt change can be heard and often sounds artificial.

To make it possible to create settings where the dynamic effects are more natural sounding, JVP incorporates a Knee control which allows you to soften the transition between sections of the gain curve. The graph below shows a curve which has “soft knees”, making the dynamic transitions more subtle.

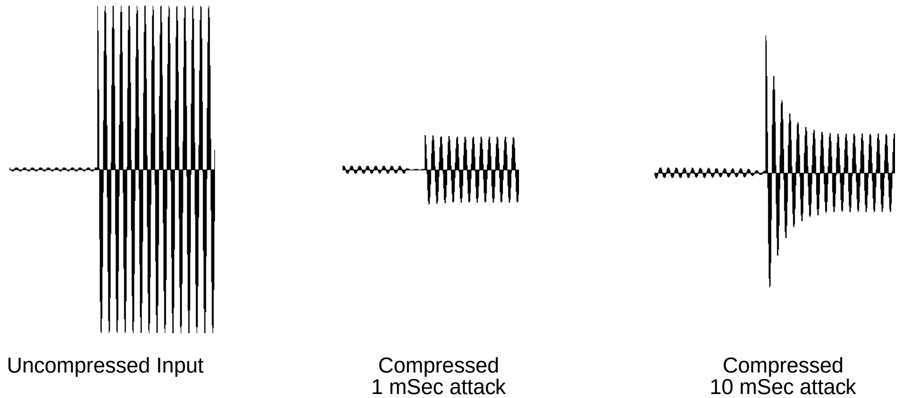


The details of operating the compressor’s Knee Control are discussed in “The Compressor/Gate” on page 36.

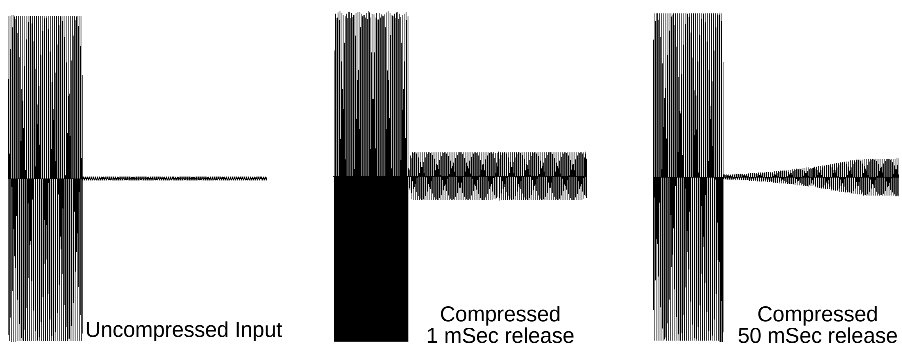
Attack And Release Times

The attack time of a compressor is simply how long it takes for the compressor to react once the input level has met or exceeded the threshold level. With a fast attack time, the signal is brought under control almost immediately, whereas a slower attack time will allow the start of a transient or a percussive sound to pass through uncompressed before the processor has time to react. Creating a deliberate overshoot by setting an attack time of several milliseconds is an effective way to emphasize the percussive nature

of instruments. The illustration below shows the effect of changing the attack time.



The release time of a compressor is the time it takes for the gain to return to normal after the input level drops below the threshold. Setting too quick a release time can cause undesirable artifacts in the output. If the release time is too long, the compressor will not accurately track level changes in the input. The illustration below shows the effect of changing the release time.

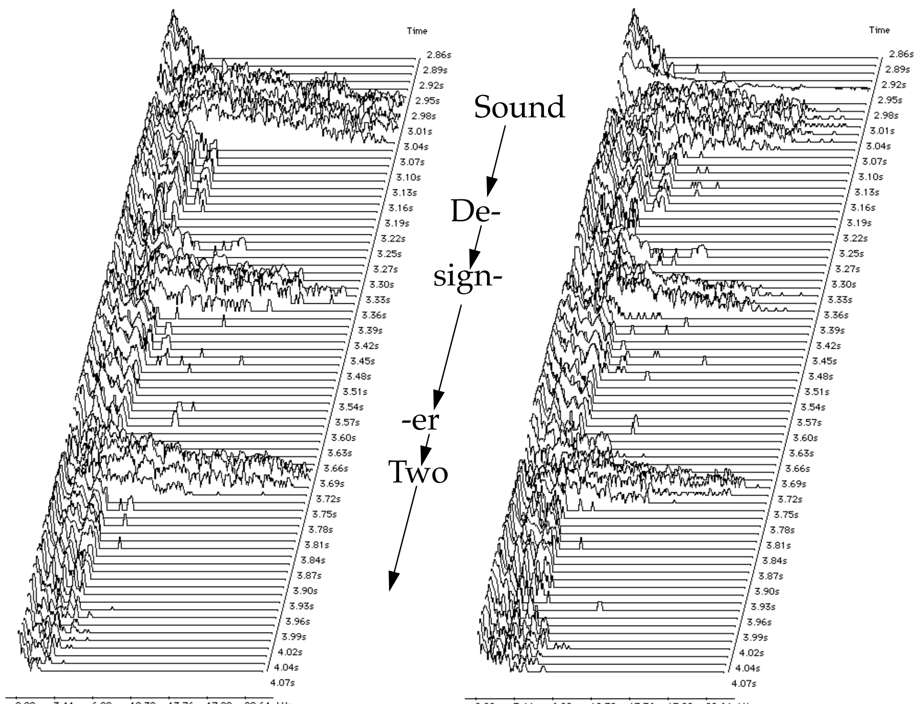


In JVP, the same attack and release times are used for both the compressor and the expander/gate.

What Is A De-Esser?

When recording spoken or sung material, the sibilants (Ss, Ts, CHs, and SHs) in the track often sound louder than the rest of the signal. The effect is unnatural and often irritating. The problem of an abnormally accentuated frequency range can also be encountered in a complete mix as well. The solution to this problem is to compress only the sibilants, thereby lowering their level relative to the rest of the track. Processing a signal this way is called de-essing.

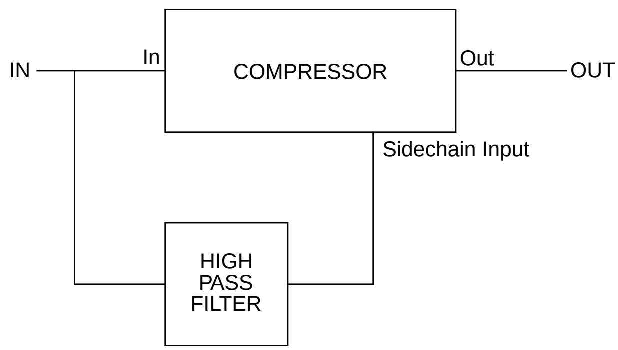
The illustration below shows an FFT display of the words “Sound Designer II” spoken by a male voice. Notice the three broadband noises that coincide with the two Ss in “Sound Designer” and the T in “two”.



Before De-essing

After De-essing

The diagram below shows how analog hardware can be configured to do the de-essing job.



Only the sibilants pass through the highpass filter. When the input signal contains sibilant material, the output of the filter causes the compressor to compress to signal. The compressor only operates when a sibilant is present.

JVP uses a digital algorithm to implement the de-esser function. While the details of the algorithm are quite complex, the diagram above illustrates the functional relationships for the purposes of explanation. The details of using the de-esser's controls are explained in "The De-esser" on page 33.

Understanding Equalization

While the subject of equalization is well understood by most people involved in digital audio, there are some facets of the technology which are not. One of these is the resolution of digital filters.

The filters used in all hardware and software equalizers have less than 16 bits dynamic range, especially at low frequencies. Typically, the last 4 bits of the sixteen bit word contain noise and distortion, generated by the mathematics of the filter design. In most applications, this noise is inaudible. It is quite audible, however, in quiet passages that contain low frequency information. Many users are unaware of this problem or have simply accepted it. The manufacturers, for obvious reasons, have not been forthcoming regarding this flaw.

The algorithms used in JVP's equalizer are specially optimized for the Motorola DSP 56000 chip used in Digidesign's DSP cards. These algorithms have *a true 140 dB* (23 bit) dynamic range. This extraordinary performance is achieved by feeding back the errors generated by the equalization process,

thereby correcting the errors to a high degree. JVP's equalizer gives performance unsurpassed by any DSP based equalizer in existence today.

Filter Types

JVP's equalizer contains five different filter types: Low Pass, Low Shelf, Peak/Notch, High Shelf, High Pass. Each filter type has its own characteristics and applications. The graphs used in the next section show the frequency response for each type with the settings used to generate the curves notated next to the graph. The details of using JVP's equalizer are covered in "The Parametric EQ" on page 40.

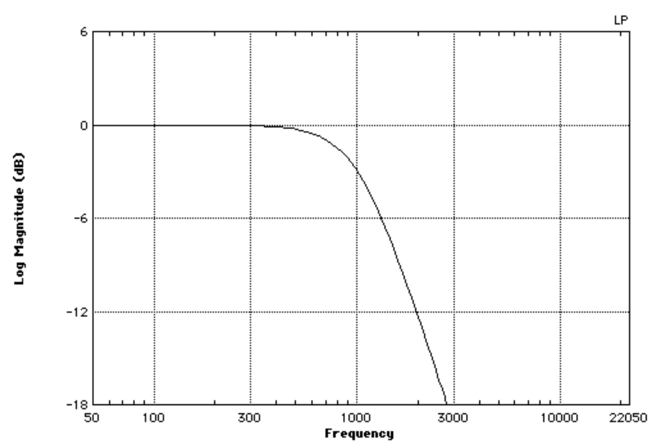
Low Pass - High Pass Filters

Low Pass Filter

Frequency: 1,000 Hz

Gain: N/A

Bandwidth: N/A



The low pass and high pass filters used in JVP have a 12 db per octave roll-off characteristic. This makes JVP very useful for attenuating sub-sonic noise, rumble, mike stand noise, high frequency hiss, and other environmental noises encountered in the recording process. Since these filters have

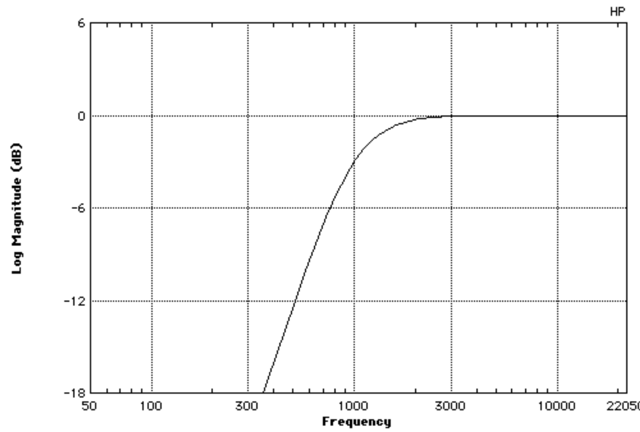
a true 140 dB dynamic range, they are especially useful for mastering stereo files.

High Pass Filter

Frequency: 1,000 Hz

Gain: + 12 dB

Bandwidth: N/A

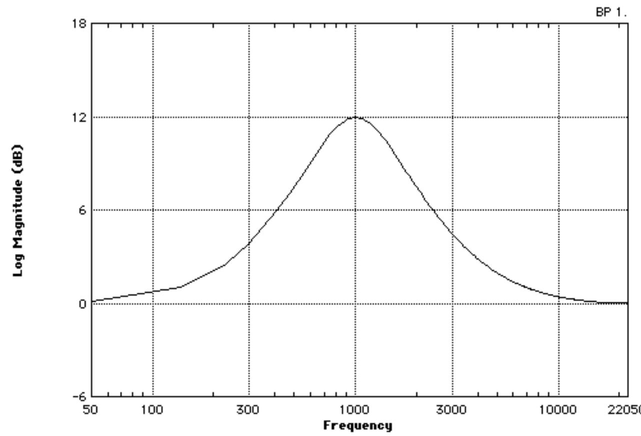


Peak/Notch Filters

JVP’s peak/notch filters are useful in shaping the tone of solo instruments and complete mixes. Beside having a 140 dB dynamic range, their control ranges are also extraordinarily broad. The range of frequencies over which the filter has effect is called the bandwidth of the filter. The bandwidth of JVP’s filters can be adjusted from between 0.1 to 4.0 octaves. The gain of the filters can be adjusted up to *±24 dB*.

JVP’s peak/notch filters can be used to shape the tone of an instrument or eliminate narrow band noises like hum. Because they can supply high gain and very narrow bandwidth, the filters can be used to accentuate the harmonics of many instrumental sounds and noises.

The graphs below show the effect of changing the bandwidth control of a peak/notch filter.

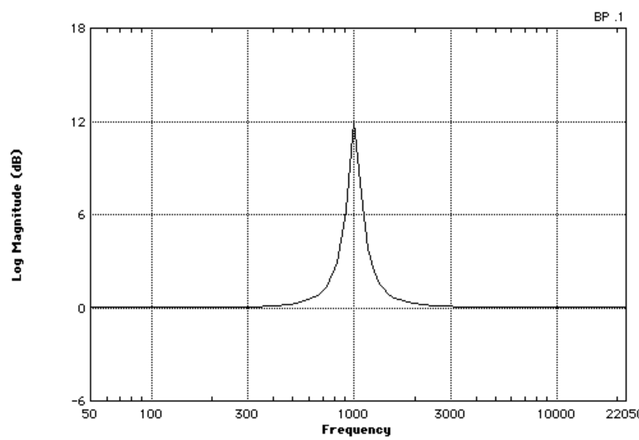


Peak/Notch Filter

Frequency: 1,000 Hz

Gain: + 12 dB

Bandwidth: 1.0 octave



Peak/Notch Filter

Frequency: 1,000 Hz

Gain: + 12 dB

Bandwidth: 0.1 octave

Note: The bandwidth of the filter is the number of octaves included between the ±3 dB points on either side of the filter's center frequency.

Shelving Filters

Shelving filters are used as primarily as "tone controls", cutting or boosting whole regions of the spectrum. A high shelf filter, for instance, acts by raising or lowering the part of the spectrum *above* the cutoff frequency. In JVP, the cutoff frequency is the frequency at which the response curve is 3 dB below/above the final gain of the filter.

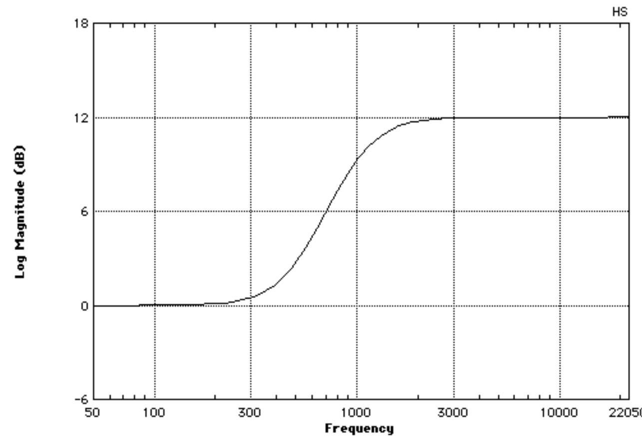
The graphs below show the response of the high shelf and low shelf filters at +12 dB gain. Notice that the slope of the roll-off is 6 dB per octave.

High Shelf Filter

Frequency: 1,000 Hz

Gain: + 12 dB

Bandwidth: N/A

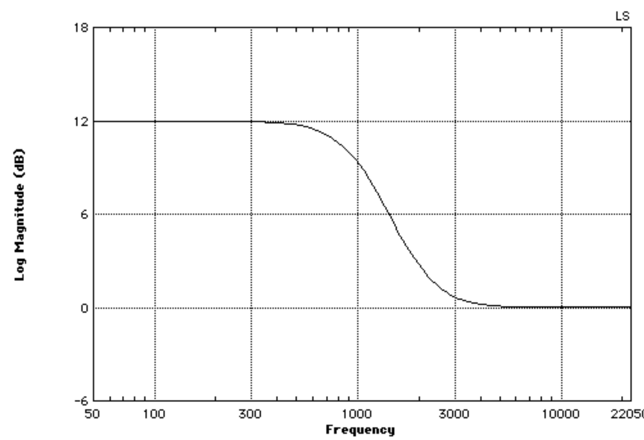


Low Shelf Filter

Frequency: 1,000 Hz

Gain: + 12 dB

Bandwidth: N/A

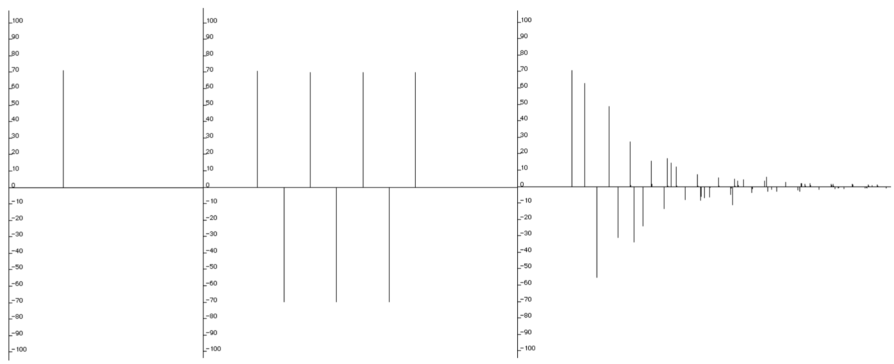


What Is A Multi-Tap Delay?

Delay effects are often used to thicken or “fatten” a solo voice or instrumental track. This thickening occurs because a copy or copies of the sound are delayed and added to the original, creating a more complex waveform. JVP contains a multi-tap delay which has *six* separately adjustable delayed signals. In addition, a comb filter with adjustable feedback is added before the

delay taps, to create even greater complexity. Using six taps, it is possible to create very complex textures from the most common material. JVP also allows you to use negative gain on the taps (the delayed signal has reversed polarity) for even more sound shaping possibilities.

The illustrations below show the effect of using a multi-tap delay. For purposes of clarity, the input signal is a simple pulse. In a musical application, a copy of the instrumental or vocal sound would begin at each point (and at the same amplitude) as the pulses seen in the graph.



Input Signal | Input + 6 taps, + and - gain | Input + 6 taps + feedback

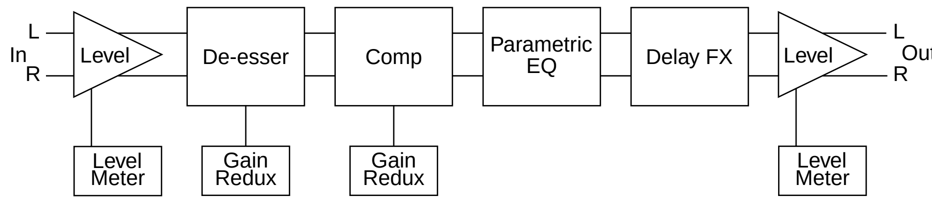
The details of using the Delay FX module are discussed in “The Delay FX” on page 44.

JVP In The TDM Environment

The TDM software environment is supported by NuBus hardware from Digidesign called the DSP Farm. Each DSP Farm card contains 4 DSP56001 chips on which JVP and other DSP plug-ins run. The TDM system requires that one chip on the DSP Farm be used for the Mixer plug-in which comes with ProTools. Assigning a large number of ProTools voices will sometimes require the use of two DSP chips on the DSP Farm.

JVP makes the most efficient use of the remaining DSP chips by running four processes on two audio channels on one chip. Normally, this would

take four DSP chips to accomplish. The illustration below shows the various DSP modules and the order in which processing occurs.

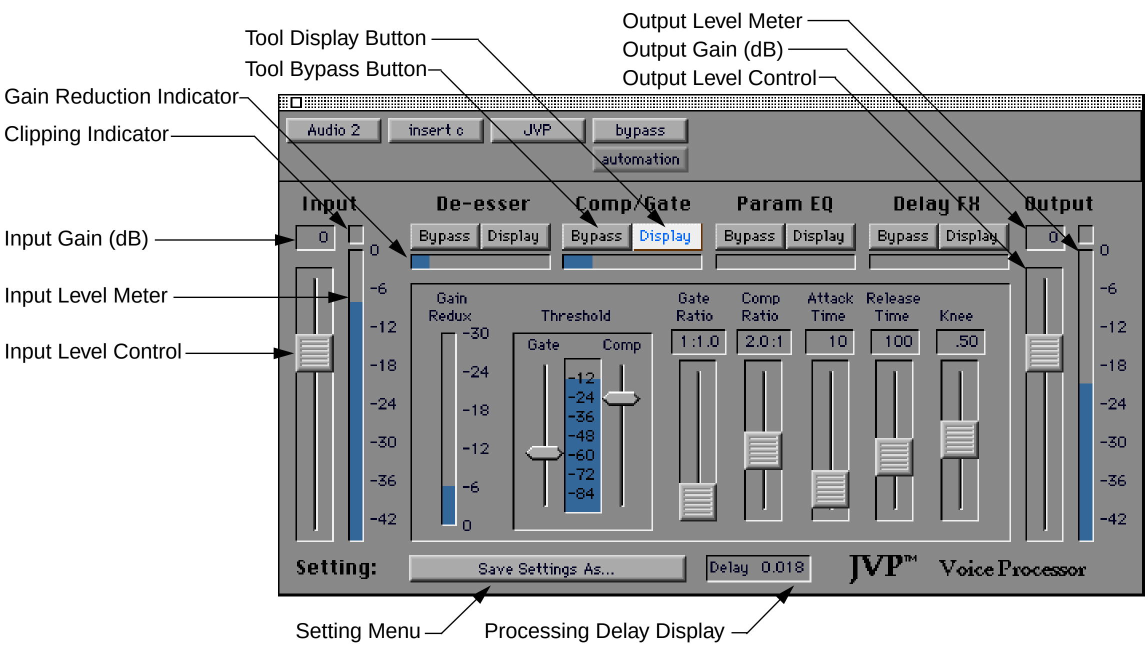


The table below shows how many DSP chips are used as more JVP channels are allocated.

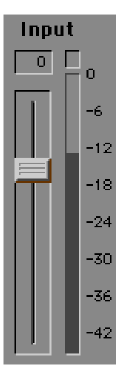
Number of JVPs used	Number of DSP chips allocated
2 mono to mono JVPs	1
2 mono to stereo JVPs	1
1 stereo to stereo JVP	1
1 mono to mono plus 1 mono to stereo JVP	2

NOTE: Sometimes TDM will post a warning dialog saying that there are an insufficient number of DSP chips available when you have added a normally legal number of JVPs. This is because of the order in which TDM allocates the DSP chips as you add more inserts. If this occurs, de-assign a JVP or two and then re-assign them. This allows TDM to sort out its DSP allocation, allowing the greatest use of the available DSP chips.

General Controls



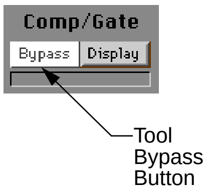
JVP contains four powerful DSP tools in one easy-to-use interface. It is configured like a normal multi-effects processor with independent input and output level controls and metering with the DSP tools connected together in series.



Input And Output Level Adjust

JVP has high resolution level meters calibrated to a resolution of 0.25 dB per pixel with a total range of 45.5 dB. For stereo files, the level displayed is the maximum of both channels. The clipping indicator becomes lit when a sound greater than or equal to -0.08 dB is input to JVP. The indicator can be lit even when the window is not visible. To reset the clipping indicator, click on it.

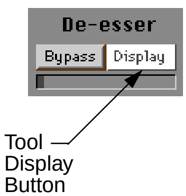
The level adjust sliders have a gain range of +24 db to -60 dB in 1 dB increments. To zero the control, press <option> and click in the area of the slot in which it travels.



Tool Bypass Buttons

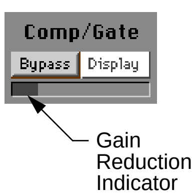
Clicking on a Tool Bypass Button removes that tool from the signal path. This is useful for isolating the effect the tool is having on the finished sound. Clicking on all the Bypass buttons will leave only the input and output level adjust sliders in the signal path.

Tool Display Buttons



The interface for each of JVP's tools is contained on that tool's control page. To display the controls for a given tool, click on that tool's Display Button. The Tool Display Buttons work like radio buttons - only one can be active at a time.

Gain Reduction Indicators



The Gain Reduction Indicators associated with the De-esser and the Compressor enable you to see these tools running when you are viewing other pages. They have a 30 dB range and display the same information as the main Gain Reduction meters on the De-esser and Compressor pages.

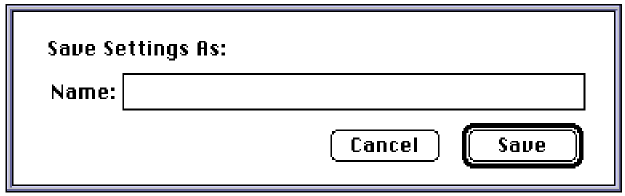
Setting Menu

JVP has a Setting menu which allows you to store your most used JVP set-ups and recall them almost instantaneously. The Setting menu allows an unlimited number of "snapshots" of JVP's settings to be saved. This data is saved in the "JVP Preferences" file in the System's "Preferences" folder and are compatible with settings from the Sound Designer version of JVP. All settings on each tool including the input and output level settings can be saved and recalled using the Setting menu.

To save settings in the Settings Menu,

1. Choose Save Settings As… from the Setting menu.

A dialog box appears requesting a setting name.



2. **Type the name of the setting in the text box and click on Save.**

   The settings are now saved under the name you typed. These settings appear at the bottom of the Setting menu.

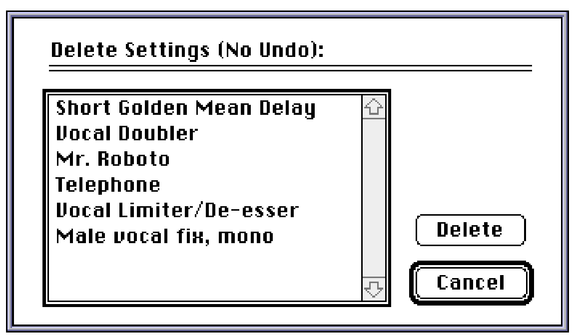
To recall the setting;

1. **Choose the desired setting from the Setting menu.**

To delete the setting;

1. **Choose Delete Settings… from the Setting menu.**

   A dialog appears with the list of the settings in the menu.



2. **Scroll down to the desired setting and select it by clicking on it.**

   You may select multiple settings for deletion by shift-clicking or click dragging over the items to be deleted. To make a non-contiguous selection, press <command> and click on the items to select them. Note that the control values of JVP remain unchanged.

3. **Click on the Delete button.**

The setting is deleted from the menu.

Processing Delay Display

The Processing Delay Display shows the amount of time in seconds that JVP will delay a track's sound output. Delay is introduced by the Compressor and the De-esser to enhance their level detection capabilities. Tracks can be slipped to the left by the amount shown in the Processing Delay Display to resynchronize their output with other non-processed tracks.

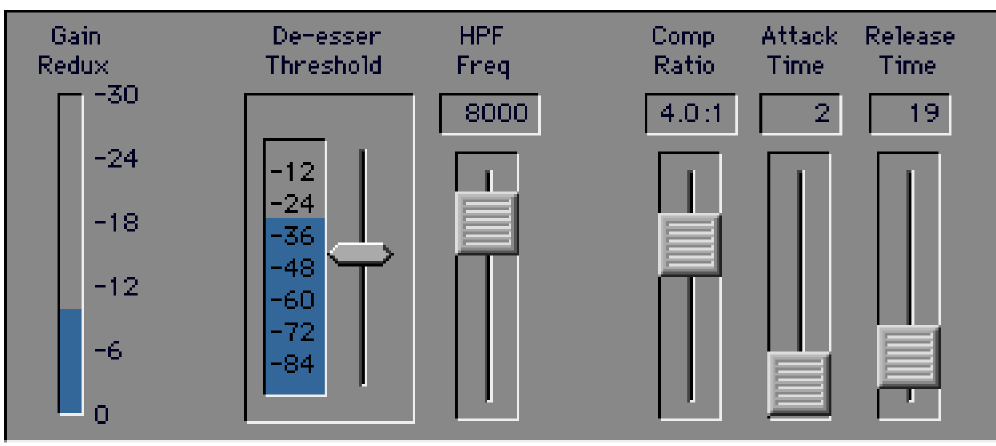
Note: The delay value displayed does not reflect additional delay introduced by the TDM system when the output is printed to a track.

CHAPTER 3

Using JVP

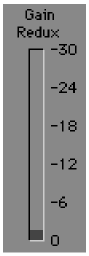
This chapter contains the details of using JVP's controls and a short how-to section on each of the tools.

The De-esser



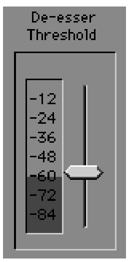
The de-esser's controls are accessed by pressing the Display Button under the word "De-esser" in JVP's display. The settings on the de-esser's controls are still active even if they are not currently displayed.

The Controls



Gain Reduction Meter

The Gain Reduction Meter displays the amount of compression of the signal. If the frequency and threshold controls are set properly, the meter will display little gain reduction during vowel sounds and soft consonants, and substantial gain reduction during sibilants.

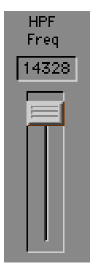


Threshold Control/Level Meter

The threshold control includes a level meter which displays the output of the de-esser's high pass filter. The control knob is adjusted by dragging it with the mouse. The effect of the change is heard while the knob is being dragged.

Frequency Control

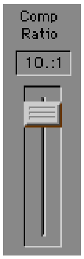
The de-esser can be viewed as a compressor which is keyed by the output of a high pass filter. When the filter passes a signal larger than the compressor's threshold, the compressor clamps down on the signal. Since sibilants are primarily broad band noise above 5 KHz, the filter is designed to pass these signals and attenuate low frequency signals as much as possible.



The filter used in JVP's de-esser is a two pole, 12 dB/octave high pass design which effectively distinguishes between sibilants and other vocal sounds. You can hear what the compressor input sounds like by using the high pass filter option in the Parametric EQ.

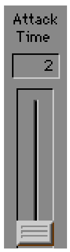
The range of the control is from 10 Hz to 20,000 Hz. The control knob is adjusted by dragging it with the mouse. The effect of moving the slider is heard during the drag. Fine adjustments can be made to the frequency value by pressing <command> before beginning the drag. The resolution is increased by a factor of ten.

Ratio Control



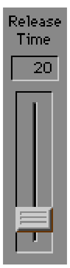
This control adjusts the compression ratio of the de-esser's compressor element. The range of the control is from 1.0 to 1 to 99 to 1. The control is adjusted by dragging the knob with the mouse.

Attack Time Control



The attack time control adjusts the speed with which the compressor element responds to peaks in the signal coming from the high pass filter. The range of the control is from 0 milliseconds to 100 milliseconds. The control is adjusted by dragging the knob with the mouse.

Release Time Control



The release time control adjusts the time it takes for the compressor element's gain to increase 6 dB after the input level drops below the threshold. The range of the control is from 0 milliseconds to 4,999 milliseconds. The control is adjusted by dragging the knob with the mouse.

Using The De-esser With ProTools

JVP's de-esser is very easy to use. The following instructions will guide you through a typical workflow.

4. **Start a ProTools session containing vocal tracks.**
5. **Select "Show Inserts View" from the Display menu.**
6. **Check "Loop Selected Region" in the Options menu.**
7. **Solo a vocal track and select a short phrase that contains sibilants.**
8. **Choose "JVP" from the insert menu of the vocal track.**
9. **Click on the Tool Display Button for the De-esser.**
10. **Click on the De-esser's Bypass button to enable it.**
11. **Start playback.**

    Notice how the sibilants increase the signal level displayed in the Threshold Level Meter.

12. Adjust the Frequency control to maximize the difference between the peaks caused by the sibilants and the other sounds in the track.

Frequency settings around 13,000 Hz work best in most cases.

13. Adjust the Threshold control so that it is just above the general level of the non-sibilant sounds in the track.

Notice that the amount of gain reduction is maximized during sibilant sounds.

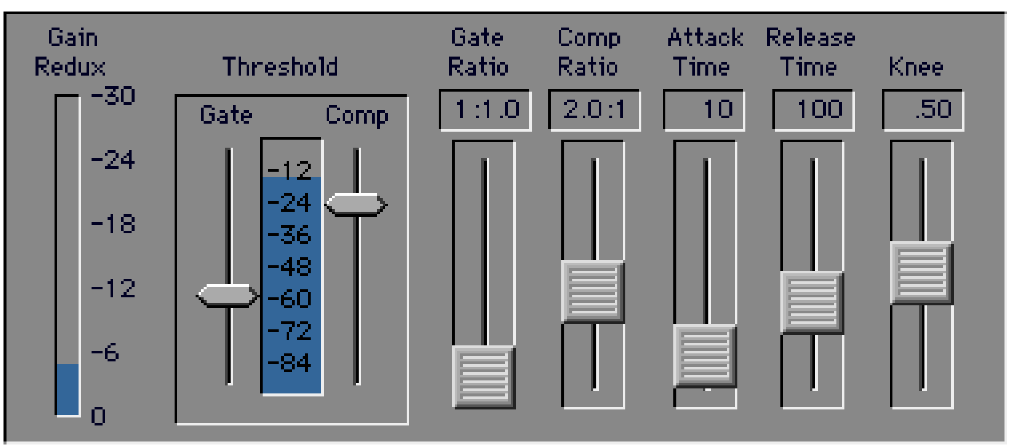
14. Adjust the compression ratio, attack and release times as desired.

Increasing the compression ratio, increases the amount of gain reduction and therefor the amount of de-essing.

15. Stop playback.

Other tools can now be added to the processing chain by clicking off their Tool Bypass Buttons.

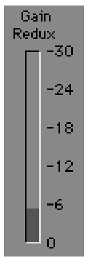
The Compressor/Gate



JVP's compressor/gate uses powerful new DSP technologies originally developed for the Multiband Dynamic Tool. While it doesn't have the flexibility of the MDT, the compressor/gate offers you all the controls found on

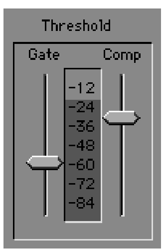
hardware compressors with the added bonus of 24 bit audio quality plus a Threshold Level Meter to take the guess work out of setting the gate and compressor thresholds.

The Controls



Gain Reduction Meter

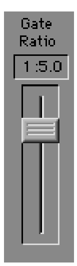
The Gain Reduction Meter displays the amount of compression of the signal. When the input signal is above the compression threshold or below the gate threshold, the Gain Reduction Meter will display the amount of compression being applied. The range of the meter is 30 dB.



Threshold Controls/Level Meter

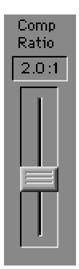
The threshold controls include a level meter which displays the actual level of the input signal that the compressor/gate uses for gain calculations. This level is influenced by the attack and release times. This can be demonstrated by observing that the apparent input level increases as the release time is increased.

The control sliders are adjusted by dragging the knob.



Gate Ratio Control

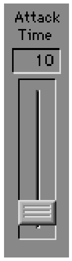
This control adjusts the expansion ratio of the downward expanding gate. The range of the control is from 1:1.0 to 1:99. The control is adjusted by dragging the knob with the mouse.



Compressor Ratio Control

This control adjusts the compression ratio of the compressor. The range of the control is from 1.0:1 to 99:1. The control is adjusted by dragging the knob with the mouse.

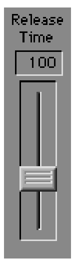
Attack Time Control



The attack time control adjusts the speed with which the compressor/gate responds to peaks in the input signal. The range of the control is from 0 milliseconds to 100 milliseconds. The control is adjusted by dragging the knob with the mouse.

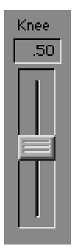
Application Note: While using the compressor and the de-esser simultaneously, make sure that the attack time of the compressor is long enough to not interact with de-esser.

Release Time Control



The release time control adjusts the time it takes for the compressor's gain to return to normal after the input level drops below the threshold. The range of the control is from 0 milliseconds to 5,000 milliseconds. The control is adjusted by dragging the knob with the mouse.

Knee Control



The knee control adjusts the sharpness of the transitions between uncompressed and compressed signals. As a signal increases in level through the threshold, the compressor will begin to apply gain reduction. With a soft knee (1.00), that change in gain will be gradual and therefor less noticeable.

The range of the control is from 0.00 (hard knee) to 1.00 (soft knee. The control is adjusted by dragging the knob with the mouse.

Using The Compressor

1. **Start a ProTools session containing vocal tracks.**
2. **Select "Show Inserts View" from the Display menu.**
3. **Check "Loop Selected Region" in the Options menu.**
4. **Select a short phrase that contains large amplitude peaks.**
5. **Choose "JVP" from the insert menu.**
6. **Click on the Tool Bypass Button for the Compressor.**
7. **Click on the Tool Display Button for the Compressor.**
8. **Start playback.**

Notice the signal level displayed in the Threshold Level Meter. Determine the average level and the peak level.

9. **Adjust the Attack Time and Release Time to best track the changes in the input signal.**

   For most spoken material, attack times between 1 and 10 mSec work well. Release times around 50 - 100 mSec. work well for material recorded in a dry, quiet environment.

10. **Adjust the Compressor Threshold control so that it is just above the average level of the track.**

    In most cases, compressing the peaks is all that is necessary. As the threshold is brought down, observe the gain reduction meter displaying the amount of compression in use.

11. **Adjust the compression ratio, attack time and release time to fine tune the sound.**

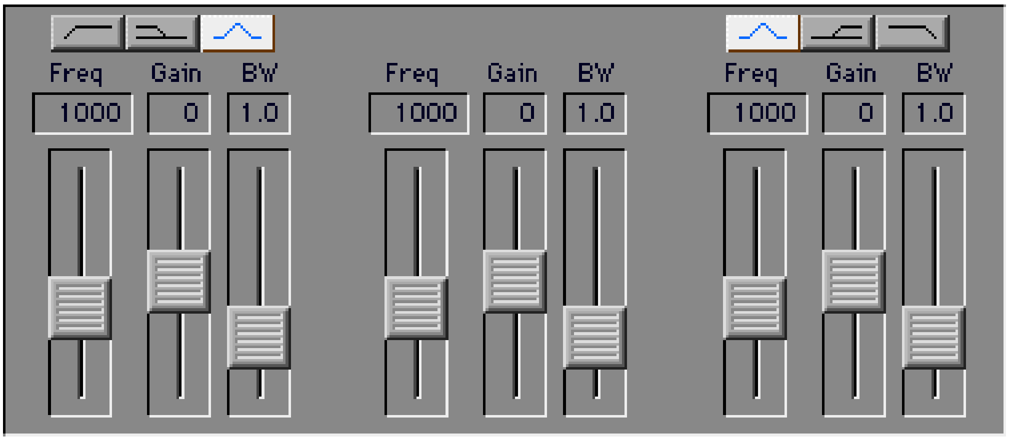
    Increasing the compression ratio, increases the amount of gain reduction and therefor the amount of compression.

12. **Adjust the Gate Threshold and Gate Ratio to gate out breath sounds, background noise, etc. in between the spoken passages.**

    The intensity of the gating effect can be adjusted using the Gate Ratio control.

13. **Stop playback.**

The Parametric EQ



JVP's parametric EQ uses powerful new algorithms optimized for the 56000 chip used in every Digidesign DSP card, giving a true 140 dB dynamic range and minimal distortion.

The EQ has three filter sections. Two of the sections can be configured as different filter types giving you a wide range of possibilities for controlling timbre and eliminating noise.

When the Parametric EQ is not bypassed, all three filters are operating continuously. The filters are designed to be completely "out of the circuit" when the Gain Control is set to 0 dB.

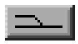
The Controls

Filter Selector Buttons

The Filter Selector buttons operate as radio buttons. That is, only one can be selected at any time. The left filter section can be a low pass, a low shelf, or a peak/notch filter. The right filter section can be a high pass, a high shelf, or a peak/notch filter. The center filter section is always a peak/notch filter.

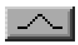
Low Pass Filter

This filter type allows energy below the cutoff frequency to pass. JVP uses a 12 dB/octave filter here to optimize for hiss reduction, psychoacoustic mixing effects, and so on.



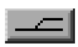
Low Shelf Filter

The low shelf filter can be used as a tone control in full mixes or to enhance the “body” of solo vocal and instrumental tracks.



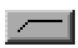
Peak/Notch Filter

The Peak/Notch Filter type is generally useful in coloring voice and instrument tracks in a variety of ways. It is often the most used filter type in parametric EQ applications.



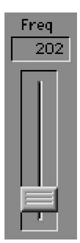
High Shelf Filter

The high shelf filter type is used as a tone control in full mixes and to add “air” to voice tracks.



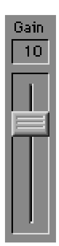
High Pass Filter

High pass filters allow energy above the cutoff frequency to pass. JVP uses a 12 dB/octave filter here to optimize for rumble reduction, sub-sonic filtering, and so on.



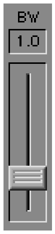
Frequency Control

The frequency control is adjusted by dragging the knob. The range of the control is from 10 Hz to 20,000 Hz. The effect of moving the slider is heard during the drag. Fine adjustments can be made to the frequency value by pressing <command> before beginning to drag the knob.



Gain Control

The gain control adjusts the amount of boost or cut given to the signal by the filter section. The range of the control is from +24 dB to -24 dB, adjustable in 1 dB increments. The gain control is adjusted by dragging the knob with the mouse. When the control is set at 0 dB gain, the filter section is effectively disabled. To zero the control, press <option> and click in the area of the slot in which it travels.



Bandwidth Control

This control is enabled only when a filter section is in peak/notch mode. The bandwidth control adjusts the width of the peak/notch filter. The range of the control is from 0.1 octaves to 4.0 octaves, adjustable in 0.1 octave increments. The control is adjusted by dragging as above.

Using The EQ

The operation of JVP's parametric EQ is similar to all other digital EQs. It is distinguished by its 24 bit dynamic range and unusual range of adjustment. The following are some tips to consider when using the parametric EQ.

Avoiding Distortion

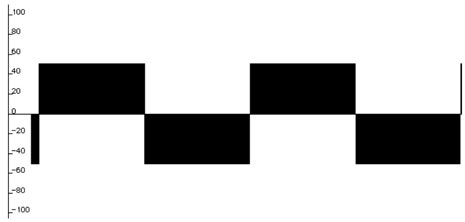
If you hear clipping in the signal, but the output level meter doesn't indicate it, you've probably caused internal clipping of the 24 bit data stream. This condition will almost always occur when the input signal is near full scale and positive gain is added by the filter sections. The remedy is to lower the input level using the input level control. Since JVP's filters are applied in order from left to right, putting a -24 dB filter before a +24 bit filter at the same frequency will avoid clipping.

Zero Phase EQ

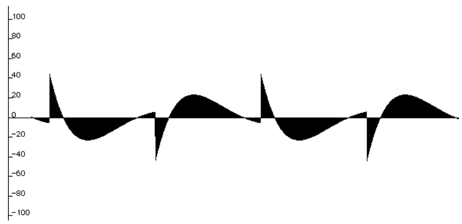
All filters, both digital and analog, introduce phase shifts or delays when they filter the signal. Phase shifts are timing delays which can effect the transient material in musical signals. These effects are heard as the blurring of stereo images and smearing of percussive transients. To eliminate phase shifts while applying EQ, follow the steps below.

1. **EQ the selected data at 1/2 the gain you intend in the final output.**
2. **Reverse the selected data in time.**
3. **EQ the data again using exactly the same settings as before.**
4. **Reverse the data again to return it to its original state.**

The effect of the EQ will be as expected except that any timing errors introduced by phase shift will be canceled.

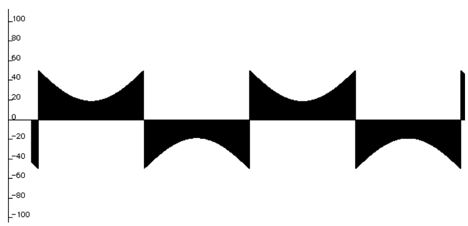


80 Hz square wave



80 Hz square wave filtered twice with an 80 Hz, 12 dB/octave high pass filter.

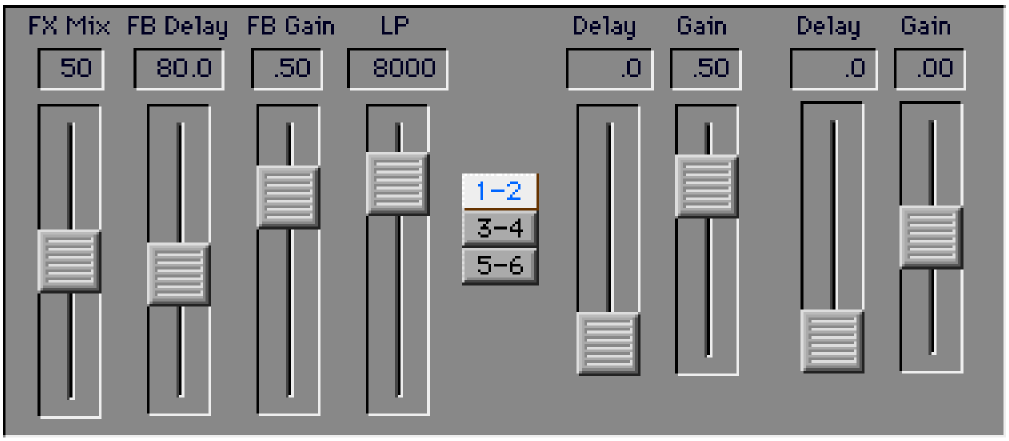
Net EQ: ~6 dB down @ 80Hz



80 Hz square wave filtered twice with an 80 Hz, 12 dB/octave high pass filter using the zero phase method.

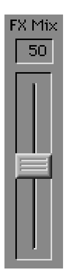
Net EQ: ~6 dB down @ 80Hz

The Delay FX



JVP's Delay FX tool is unusual because it provides more control than the delays normally found in hardware reverb and delay units. In addition to having six individually adjustable taps, the Delay FX tool allows you to use negative gain and adjustable comb filtering to create a wide variety of musical results. Its controls are familiar and easy to use, so creating complex delay effects from scratch can be done by ear.

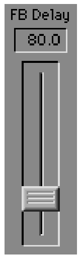
The Controls



FX Mix Control

The FX Mix control adjusts the balance between the input signal and the signal generated by the delay. The range of the control is from 0 to 100, zero being 100% input signal. The control is adjusted by dragging the knob.

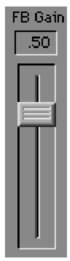
For a stereo to stereo JVP, the delay effects are added into the stereo mix using the Pan Controls mentioned below. In a mono to stereo JVP, the delay effects are added via the Pan Controls to the mono signal to produce a stereo output. In a mono to mono JVP, the delay effects are added directly to the mono input signal and produce a mono output.



Feedback Delay Control

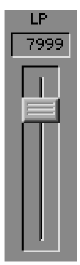
This control adjusts the amount of delay used in the comb filter. The range of the control is from 0 milliseconds to 371.3 milliseconds. Large delay values will give an echo-like effect. Small delay values will thicken the sound or, if the Feedback Gain value is high, create a pitched effect.

Feedback Gain Control



The Feedback Gain control adjusts the amount of signal that is fed back through the comb filter. The range of the control is from -99% to +99%. Negative gain settings reverse the polarity of the signal being fed back. The control is adjusted by dragging the knob.

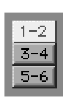
When the control is set at 0 dB gain, the feedback section is effectively disabled. To zero the control, press <option> and click in the area of the slot in which it travels.



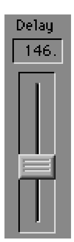
Low Pass Filter Control

This control adjusts the amount of high frequencies that are used in the feedback signal. The range of the control is from 10 Hz to 20,000 Hz.

Delay Tap Display Buttons



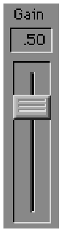
Because of limited screen turf, only two delay taps are displayed at one time. Pressing on a Delay Tap Display Button will cause the associated taps to appear on the Delay FX page. Only one pair of taps can be displayed at a given time.



Delay Time Control

This control adjusts the amount of delay the signal receives through the associated tap. The range of the control is from 0 milliseconds to 371.4 milliseconds.

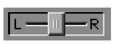
Gain Control



The delay tap's gain control adjusts the amount of signal that is contributed to the delay FX mix by the tap. The range of the control is from -99% to +99%. The control is adjusted by dragging the knob.

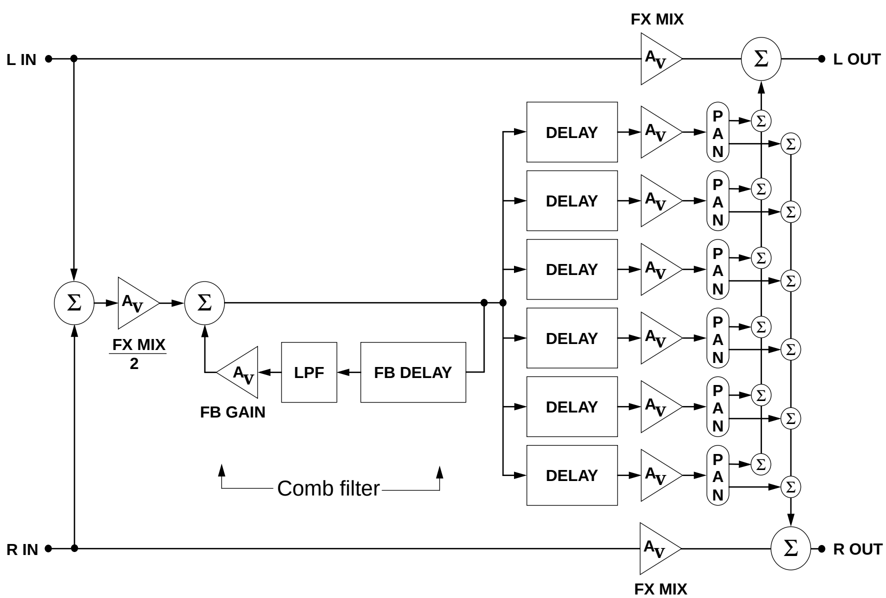
When the control is set at 0 dB gain, the delay tap is effectively disabled.

Pan Control



In a stereo to stereo or mono to stereo JVP, the output of each delay tap can be panned to any location between the Left and Right channels. The control is adjusted by dragging the slider with the mouse or by pressing on or about the slot in which it travels. The effect of moving the slider is heard during the drag. This control is disabled when processing a single channel or mono.

Using The Delay



The diagram above shows the signal flows inside JVP's Delay FX module. Notice that the two stereo channels are summed before being input into the

delay. The signal then flows through the comb filter. The function of this element is to “echo” the sound. This echoing causes periodic notches in the frequency response of the output. Hence the name comb filter. Long feedback delay times will actually sound like real echoes.

The signal then goes through one or many of the delay taps. Each tap can be individually adjusted for delay and gain. Using a negative gain setting, reverses the polarity of the delayed signal, which makes for complex interactions in the frequency domain when using short delay times. Finally, the delayed signals are panned and summed to the outputs.

Using JVP’s Delay FX module is easy because the controls are familiar, readily accessible, and can be adjusted by ear. Here are a few tips which will prove useful when using JVP’s Delay FX.

No Output?

At least one of the taps must be “turned on” for the delayed signal to be heard. (Refer to the diagram.) Remember that the taps are turned off when their gain is zero.

Adding Pre-Delay

The output of the comb filter section can be delayed by using a tap. The delay time of the tap is equal to the “pre-delay” time you want.

Simulating “Early Reflections”

The six delay taps can be configured to simulate early reflections by keeping the delay times under 50 milliseconds. Sound travels at about one foot per millisecond. If you are standing twenty feet away from a wall, the sound you make will echo back to you 2 times 20 milliseconds or 40 milliseconds later. Using the delay times of the taps, you can design the early reflections of any rectangular room from a closet to a cathedral!

Index